COMPUTATIONALLY EFFICIENT HAMMOND ORGAN SYNTHESIS

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ABSTRACT

The Hammond organ is an early electronic musical instrument, which was popular in the 1960s and 1970s. This paper proposes computationally efficient models for the Hammond organ and its rotating speaker system, the Leslie. Organ tones are generated using additive synthesis with appropriate features, such as a typical fast attack and decay envelope for the weighted sum of the harmonics and a small amplitude modulation simulating the construction inaccuracies of tone wheels. The key click is realized by adding the sixth harmonic modulated by an additional envelope to the original organ tone. For the Leslie speaker modeling we propose a new approach, which is based on time-varying spectral delay filters producing the Doppler effect. The resulting virtual organ, which is conceptually easy, has a pleasing sound and is computationally efficient to implement.

1. INTRODUCTION

The Hammond organ was one of the very first electronic synthesizers constructed in the early 20th century. Originally it was designed to be a low-cost substitute to the pipe organs used in churches, but later in the 1960s and 1970s it also became a commonly used keyboard instrument in popular music productions due to its characteristic timbre. This special timbre of the Hammond organ was partly produced by the speaker that was used for the reproduction of the organ sound. This reproduction device, the Leslie speaker, had two rotating units and its original target was to simulate the continuously shifting sound sources emanating from the different parts of the broad wall of pipes of the church organs.

Due to the difficulties in transporting the heavy organ unit (almost 200 kilograms) and the Leslie speaker and the possible technical failures of the electromechanical instrument, musicians have craved for a more easily portable unit that can be operated reliably. To meet this demand, many electronic and digital keyboards have a readily available tone that imitates the sound of the Hammond organ. On the other hand, the sound of other organs that did not necessarily imitate the Hammond organ have also been emulated, see e.g. [1]. In addition, the effect produced by the Leslie speaker is still widely used and devices that emulate the processing performed by the speaker are also available.

In this paper, computationally efficient models for both the Hammond organ and the Leslie speaker are presented. The sound generation principle used in the organ is presented Section 2 and its discrete-time replication is discussed. Section 3 presents the proposed Leslie rotating speaker effect model that is based on time-varying spectral delay filters (SDFs). The computational complexity of the models is discussed in Section 4. Section 5 concludes the paper.

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2. ORGAN MODEL

The Hammond organ was originally designed to be a more compact version of the Telharmonium synthesizer [2]. Similarly to the Telharmonium, the Hammond organ mixed the sound of several electrical tone generators, tone wheels, to synthesize a desired sound [3]. Each tone wheel consisted of a metallic disk having a grooved rim in the proximity of a magnetic pickup. When the disk was rotated, the grooves of the rim induced an alternating current to the coil of the pickup. The generated current was sinusoid-like, and the frequency of the sinusoid depended on the number of grooves and the speed of rotation [3].

For each key in the manual (in practice, a keyboard) seven tone wheels were assigned and they produced the first, the second, the third, the fourth, the fifth, the sixth, and the eighth harmonic of the frequency corresponding to the musical note associated with the key [3]. Since there were only a limited number of tone wheels built inside the organ chassis, the produced harmonics were not always exact multiples of the fundamental frequency. In such cases, the tone wheel that produced the closest frequency to the harmonic was selected [3]. In the later production models of the Hammond organ, the number of tone wheels associated with a key was increased to nine. The added components were the octave below the fundamental frequency (0.5 times the fundamental) and the musical fifth (1.5 times the fundamental).

The amplitudes of the harmonics were controlled separately with a set of sliding controllers, drawbars [3]. The original organ design contained two hand-operated manuals for both of which an individual drawbar was associated. In addition, the original design contained a pedal manual that had a separate drawbar with only two contollers [3]. In the later production models, the drawbar of the pedal manual was merged with the other drawbars to create two nine-bar controllers. In the early models, the drawbars had discrete values ranging from zero (off) to eight (fully on). Later, continuous-value controllers were introduced to provide a more flexible control. In addition to the manual mixing of the generated harmonics, the original design of the Hammond organ included a set of preset switches that set the amplitudes of the harmonics to predefined values [3].

The Hammond organ also contained a tremolo effect unit that could be applied to the resulting sound after the mixing stage [3].

The frequency of the tremolo effect, five Hertz, was set by the synchronous motor rotating the tone wheels. The depth of the tremolo effect could be controlled continuously, and the effect was more pronounced at high frequencies than at low frequencies due to the construction of the effect [3].

2.1. Digital modeling of the Hammond organ

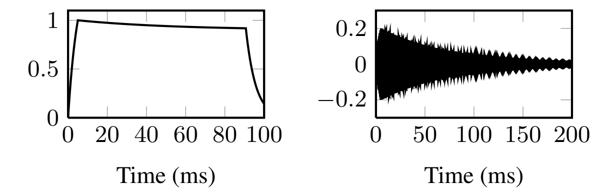
According to the description given above, a simplified digital Hammond organ model can be constructed using the additive synthesis technique. The number of sinusoids is directly set by the number of tone wheels dedicated for a key. However, it should be noted that since the amplitude of the sinusoids is controlled by the drawbar, some of the sinusoids can have an amplitude that is so weak that they are not reasonable to be synthesized at all. In addition to the sinusoids required for the actual organ tone synthesis, a sinusoid with frequency of five Hertz can be included to the model to simulate the tremolo effect.

Since the drawbars were used to control the timbre of the produced sound by setting the individual amplitudes, there are numerous possible combinations even when the discrete-valued drawbars are used. Of these possible combinations, a mixture that consists of four components is considered as an example timbre in this paper. The (exact) harmonic components used in the example tone are the first, the second, the third, and the eighth when the fundamental frequency is above 130 Hz (note C3). Below that frequency the first and the eighth harmonic are used together with the sub-octave harmonic and the musical fifth. The notes above the given frequency can be understood to be played with one (hand-operated) manual whereas the notes below that frequency are played with another manual and/or with the pedal manual. These components were manually chosen to produce a pleasent tone with the manually chosen mixing setup given below.

The component amplitudes of the example timbre are 1 (full amplitude), 0.2, 0.2, and 0.1, ordered from the lowest frequency to the highest in both cases. As the general amplitude envelope an envelope that has a rapid (order of a few milliseconds) attack and release is used. After the attack phase of the amplitude envelope, the envelope can have a fast decay (a constant decay time of 50 ms) to a sustain level that is quite close to the maximum amplitude. With this decay simulation, the initial burst of current at the synthesizer output introduced by the rapid pressing of a key [3] can be simulated. An example of the general amplitude envelope is plotted in Figure 1(a).

2.2. Model extensions

The model described above can be extended in a couple of ways. First, the noise-like signal produced by the closing and the opening of the tone generation circuits at the events of key pressing and release, respectively, can be added to the generated sound at the respective events. The noise produced by these events has most of its energy at high frequencies, and especially the ”key click“ produced by the key pressing is quite pronounced and it is often used by musicians due to its percussive effect it adds to the tone. However, instead of simulating the key click by filtering a noise signal, it can be approximated by adding the sixth harmonic component with an additional amplitude envelope to the output signal. The amplitude envelope of the click signal has again a rapid attack and release (the same values as with the main tone can be used) and a fast decay (a constant decay time of 70 ms) to the zero level,



(a) (b)

Figure 1: (a) Amplitude envelope of the example timbre. (b) Time-domain plot of the proposed ”key click“ effect.

as indicated in the time-domain plot shown in Figure 1(b). This manually chosen approximation effectively adds a click-like component to the tone and it is computationally efficient to implement.

Second, since the tone wheels were mechanical devices, the rotating disks cannot be constructed to be perfectly symmetric. These imperfections in the symmetry can be included to the model with a small sinusoidal amplitude modulation (at the frequency of the tone wheel rotation and the tremolo effect, five Hertz) that simulates the small distance fluctuations between the disk and the pickup. Similarly, the imperfections in the grooving of the disk rim will produce the harmonics of the nominal frequency of the wheel.

3. ROTATING SPEAKER EFFECT

The Leslie rotating speaker is constructed from two separate rotating units, the treble unit and the bass unit [4]. The treble unit is formed by a symmetrical dual-horn structure which is rotated with a motor. Only one of these horns is used for the sound reproduction with the other acting as a dummy for symmetrical mass and form. These horns are quite directive and thus the output effect contains significant amplitude modulation. Furthermore, the location of the sound source is at the mouth of the rotating horn. This means that the distance of the horn from the listener is changing resulting in a frequency modulation effect.

The bass unit is constructed with a stationary loudspeaker with produced sound fed through a rotating wooden drum. This drum effectively adds a directional pattern to the bass sound and thus generates amplitude modulation when rotated. Henricksen [4] also mentions that it is possible that there is a frequency modulation effect present near the crossover frequency of the two units although the amplitude modulation is the dominant effect.

To create a similar effect compared to the rotating speaker both frequency modulation and amplitude modulation should be applied similarly to the real speaker. Amplitude modulation is simple to implement if the frequency-dependent directivity of the loudspeaker is dismissed. Then it is enough to multiply the signal with a sinusoidal oscillator scaled appropriately for the desired depth of effect.

The frequency modulation of an arbitrary signal can be produced by feeding the signal into a delay line and by modulating the delay-line length [5]. However, in order to produce a smooth effect, fractional-delay filters (like the linear interpolator) [6] have to be applied, which makes the implementation more complex. An alternative efficient solution is to apply spectral delay filters (SDF) for this purpose.

Several studies have previously been published about the modeling of the Leslie rotating speaker. Smith et al. [7] simulated the Doppler effect with interpolated delay lines and used Leslie simulation as an example case. Their implementation is comparable to the one presented here. Kronland-Martinet and Voinier [8] provided a study on the perceptual simulation of moving sources and, similarly to Smith et al., used the Leslie speaker as one of their application examples. Herrera et al. [9] approached the task by measuring impulse responses in a dense rotational pattern for both units. They synthesized the effect by applying a time-varying FIR filter.

3.1. Spectral delay filters

Spectral delay filters [10] are a recently proposed method for producing a frequency-dependent delay with a first-order allpass filter chain. One of their advantages is that the distribution of the delay in frequency can be controlled with the filter coefficient. The spectral delay filter is formulated by starting from a first-order allpass filter with transfer function

$$H(z) = \frac{a_1 + z^{-1}}{1 + a_1 z^{-1}}, \tag{1}$$

where $a_1 \in [-1, 1]$ is the filter coefficient.

The magnitude response of this filter is 0 dB at all frequencies. The phase response is nonlinear (unless $a_1 = 0$) and thus the group delay depends on frequency. The group delay of a single allpass section can be calculated with

$$\tau_g(\omega) = \frac{1 - a_1^2}{1 + 2a_1 \cos\omega + a_1^2}. \tag{2}$$

With absolute values of filter coefficient a_1 close to unity, the delay can be quite large at some frequencies. For example, with $a_1 = -0.9$, the delay at low frequencies is close to 19 samples.

As SDF is a cascade of first-order allpass filters, the transfer function an SDF is

$$H(z) = \left(\frac{a_1 + z^{-1}}{1 + a_1 z^{-1}}\right)^N, \tag{3}$$

where N is the length of the filter cascade. This effectively multiplies the amount of produced delay while keeping the control of delay distribution simple.

Pekonen *et al.* [11] and Kleimola *et al.* [12] studied the possibility of varying the filter coefficient of an SDF through time. It was noted that the stability conditions for time-varying SDFs are the same as with the single allpass filter, i.e. $|a_1| \leq 1$, if the condition holds for every time step n [11]. Here, the difference equation of the Direct form I realization,

$$y(n) = m(n)x(n) + x(n-1) - m(n)y(n-1), \tag{4}$$

where $x(n)$ is the input signal, $y(n)$ is the output signal, and $m(n)$ is the modulator signal, is used [13, 12].

The modulator signal, and how it is applied, can significantly affect the produced effect. For example, a sinusoidal modulator produces different effects if it varies between -1 and 1 or between -1 and -0.9. Thus, for this paper, two additional variables are defined for controlling the modulation. This is done by performing the substitution

$$m(n) = M_{\mathrm{s}} m_0(n) + M_{\mathrm{b}} \tag{5}$$

to (4). Here, $m_0(n) \in [-1, 1]$ is the unscaled modulator signal, M_{s} is a modulator scaling term, and M_{b} is a modulator bias term.

3.2. Implementation

Based on the construction of the Leslie speaker, a model for implementation can be created. A block diagram of this model is shown in Figure 2.

The modulating signal should be sinusoidal as that represents the movement of the real speaker best. As there are separate motors for the treble unit and the bass unit in the real speaker, it is prudent to also use two separate oscillators for modulation. This enables the use of the characteristic speed-up and slowdown effects of the Leslie speaker where the bass unit accelerates slower due to the inertia of the unit. This acceleration mismatch is simple to implement with different modulator frequency envelopes.

The highpass and lowpass filters can be created with any suitable filter implementation although a digital version of the real crossover filters would be preferable. In the demonstration implementation, these filters were fourth-order digital Butterworth IIR filters with the cutoff frequency at 800 Hz.

Amplitude modulation can be implemented directly by multiplication with a scaled modulator. This scaling is done so that the values of the sinusoid are between one and α, where α is a value between 0 and 1.0. In the demonstration implementation, α was selected to be 0.9 for both paths.

The frequency modulation is performed with two SDFs. The parameters of the SDFs differ between the treble and the bass pathways to optimize the effect on both pathways. The parameters presented in Table 1 were found to produce a pleasant effect.

The modulator frequency f_m for slow rotation speed was 2 Hz, and for the fast speed was 6 Hz. The modulator of the treble pathway had 0.1 Hz higher frequency compared to the bass pathway. This difference is introduced to model the motors running at similar but slightly different speeds, as described by Henricksen [4] mentioned that the speeds of the real motors are almost but not exactly the same. It was found that the effect is slightly more interesting when there is this mistuning present in the modulator frequencies.

4. DISCUSSION

The organ model presented in Section 2 provides an imitation of the original Hammond organ design. In general, the computational load of the model is rather low as it requires only ten sinusoidal oscillators when each of the nine tone components are generated separately. The tenth oscillator is dedicated for the tremolo effect. If the key click is included, one can use the output of the oscillator that produces the sixth harmonic. For the example timbre described in Section 2, only seven oscillators are required in total. It should be noted, however, that the actual computational load, i.e. the number of cycles used by the processor, depends on how the sinusoidal oscillators are implemented and that the different sine oscillator implementations may be computationally more costly in some processors than in other processors.

The number of oscillators can obviously be decreased by tabulating a predefined mixture of sinusoids into a wavetable. However, since there are numerous possible mixtures, the memory consumption of the wavetable-based approach would be huge. Therefore, for the general Hammond organ model the tones are generated more efficiently by synthesizing each of the sinusoids separately. Yet, for a small set of predefined mixtures that are not modified afterwards, like in the case of the example timbre of this

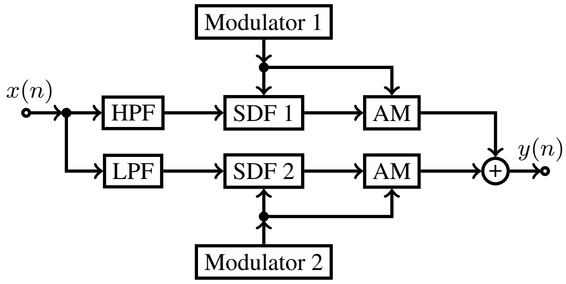


Figure 2: Block diagram of the rotator effect.

Table 1: Parameter values for the SDFs in Figure 2.

Parameter	Treble (SDF1)	Bass (SDF2)
SDF length N	4	3
Modulator scaler M_{s}	0.2	0.04
Modulator bias M_{b}	-0.75	-0.92

paper, the wavetable-based approach does provide a computationally efficient implementation.

The Leslie model presented in Section 3 produces a perceptually pleasant rotator effect. However, the model is not complete and could be extended in various ways, e.g. modeling the cabinet and the loudspeaker elements, and low-complexity methods should be implemented for them in the future. Nonetheless, the advantages of this implementation are evident. The computational complexity is quite low as the filters are of low order and otherwise there are only a few required operations. Furthermore, the implementation is simple as the same oscillators can directly modulate the amplitude of the signal and the filter coefficients of the SDFs. Only proper scaling is required to produce the desired effect.

5. CONCLUSION

A computationally efficient model of the Hammond organ and the Leslie rotating speaker were presented. The original design of the Hammond organ was described and its digital implementation was discussed. A practical example setup for the organ timbre was given, and extensions to the model were discussed. A perception-based model of the Leslie rotating speaker was proposed based on time-varying spectral delay filters (SDFs). Practical parameters for the SDFs were presented and modeling of the two operating speeds of the original speaker was discussed. In addition, the computational load of both the Hammond organ model and the Leslie rotating speaker was discussed and analyzed.

As stated above, the proposed digital Hammond organ model is a simplification of the original electromechanical device. The model neglects the crosstalk, or tone leakage, of the tone wheels to their neighboring pickups. This feature can be included by adding the sinusoids generated by the neighboring tone wheels with relatively small amplitudes to the tone wheel output signal (that is, the sinusoid). However, it should be noted that the frequencies of the crosstalk components depend on the ordering of the tone wheels, and therefore no general rules can be given for the tone leakage modeling. Furthermore, this feature can increase the computational complexity of the algorithm quite much and was therefore not included in the proposed model in the first place. In addition, the proposed organ model does not include the imperfections in the disk rim grooving.

Sound examples of the proposed organ and speaker models can be found online at `http://www.acoustics.hut.fi/go/dafx11-hammond/`.

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